What is claimed is:

(1) A digital filter comprising a series of digitized time coefficients stored in a memory, said time coefficients being mapped to a like number of frequency coefficients, said frequency coefficients being spaced at frequency intervals, having either zero phase angles or linearly spaced phase angles and having amplitudes which are mirrored about a mid frequency to produce periodicity of a time response for said digital filter.

(2) A digital filter according to claim 1, wherein said time coefficients are odd in number.

(3) A digital filter according to claim 1, wherein the number of said time coefficients is equal to or greater than 5.

(4) A digital filter according to claim 3, wherein the number of said time coefficients is one of 7, 9 and 11.

(5) A digital filter according to claim 1, wherein said time coefficients are defined by inverse discrete Fourier transforms of said frequency coefficients.

(6) A digital filter according to claim 1, wherein a portion of said frequency coefficients having frequencies within a free band are selected so as to achieve a generally constant oscillation frequency across a center band which is broader than said free band.

(7) A digital filter according to claim 1, wherein said frequency coefficients are spaced at equal frequency intervals.

(8) A digital filter according to claim 1, wherein said time coefficients are integer numbers.

(9) A method of making a digital filter comprising the steps of:

establishing a plurality of frequency response coefficients separated at frequency intervals, said frequency response coefficients being graphically characterized by a base point, a series of principal points and a series of mirror points and having either zero phase angles or linearly spaced phase angles, said base point having a frequency of zero, a portion of said principal points being situated at frequencies encompassing the range of human hearing and a portion of said mirror points having frequencies and amplitudes which mirror said portion of said principal points when viewed relative to a mid frequency higher than said frequencies of said principal points,

performing inverse discrete Fourier transformations to map said frequency response coefficients into corresponding time response coefficients, and

storing said time response coefficients in a digital memory.

(10) A method according to claim 9, wherein said portion of said principal points are situated at predetermined frequencies within the range of human hearing and have amplitudes that roughly are inversely corresponding to human hearing sensitivity at said predetermined frequencies.

(11) A method according to claim 9, wherein said frequency response coefficients are established at uniformly spaced frequency intervals.

(12) A method according to claim 9, wherein said establishing and performing steps comprise the following steps:

selecting a plurality of first frequency response coefficients separated at uniformly spaced frequency intervals, said first frequency response coefficients having either zero phase angles or linearly spaced phase angles and each first frequency response coefficient further having an amplitude and a frequency,

arranging said plurality of first frequency response coefficients in order from lowest frequency to highest frequency to define a list of first frequency response coefficients,

performing inverse discrete Fourier transformations to map said plurality of first frequency response coefficients into corresponding first time response coefficients,

discarding a pair of said first time response coefficients which have equal magnitudes and are positioned adjacent to one another in said list, remaining time response coefficients defining second time response coefficients,

assessing the effect on a frequency response of the digital filter after discarding said pair of said first time response coefficients,

repeating said performing, discarding and assessing steps until a pair of discarded time response coefficients cause a significant change in the frequency response of the digital filter, and

adding to remaining time response coefficients said pair of discarded time response coefficients causing a significant change in the frequency response of the digital filter, said added and remaining time response coefficients comprising final time response coefficients.

(13) A method according to claim 12, further comprising the steps of:

multiplying each of said final time response coefficients by an integer conversion number to define converted final time response coefficients, said conversion number being sufficiently large to permit discarding any remaining fractional portion without losing substantial final time response coefficient accuracy, and

discarding from each of said converted final time response coefficients any remaining fractional portion.

(14) A method according to claim 13, wherein said conversion number is selected as a power of two.

(15) A method according to claim 12, wherein said assessing step comprises the steps of:

generating a first frequency response curve from said first frequency response coefficients,

performing discrete Fourier transformations to map said second time response coefficients into corresponding second frequency response coefficients,

generating a second frequency response curve from said second frequency response coefficients, and

comparing said first and second frequency response curves to determine if said second frequency response curve is substantially different from said first frequency response curve.

(16) A method according to claim 9, wherein each of said frequency response coefficients having an amplitude and a frequency and said range of human hearing being within a band of frequencies having a low end and a high end, a portion of said frequency response coefficients having frequencies between a reference frequency and said high end increase in amplitude as per increasing frequencies from said reference frequency toward said high end.

(17) A method according to claim 16, wherein said reference frequency is in the range of from about 501 Hz to about 8018 Hz.

(18) A method according to claim 16, wherein said frequency response coefficients having frequencies between said reference frequency and said high end increase in amplitude up to a significant amplitude peak at a peak high frequency and decrease in amplitude as per increasing frequencies toward said high end above said peak high frequency.

(19) A method according to claim 18, wherein said peak high frequency is in the range of from about 1002 Hz to about 20045 Hz.

(20) A method according to claim 18, wherein the amplification of the frequency response coefficient at said peak high frequency is from about 1.3 times to about 6.0 times the amplification of said frequency response coefficient at said reference frequency.

(21) A method according to claim 16, wherein said frequency response coefficients having frequencies between said reference frequency and said high end increase in amplitude up to a significant amplitude peak at a peak high frequency, decrease in amplitude as per increasing frequencies down to a significant amplitude trough at a trough high frequency and increase in amplitude as per increasing frequencies toward said high end.

(22) A method according to claim 9, wherein each of said frequency response coefficients having an amplitude and a frequency and said range of human hearing being within a band of frequencies having a low end and a high end, a portion of said frequency response coefficients having frequencies between a reference frequency and said low end increase in amplitude as per decreasing frequencies from said reference frequency toward said low end.

(23) A method of enhancing a series of digital audio samples comprising the steps of:

receiving said series of digital audio samples, and

generating a driving signal by convolving said series of samples in real time with a series of stored time coefficients, said time coefficients being mapped to a like number of frequency coefficients, said frequency coefficients being spaced at frequency intervals, having either zero phase angles or linearly spaced phase angles and having amplitudes which are mirrored about a mid frequency.

(24) A method as set forth in claim 23, further comprising the step of generating an analog audio signal from said driving signal.

(25) A method according to claim 23, wherein said time coefficients are integer time coefficients.

(26) A method according to claim 23, wherein said step of generating a driving signal comprises the step of:

repeatedly solving the following equation for Y:

$$Y(n) = A_0X(n) + A_1X(n-1) + \dots + A_{N-1}X(n-[N-1])$$

where A_0 through A_{N-1} are said stored time coefficients;

$X(n)$ is the most recent sample received;

$X(n-1)$ through $X(n-[N-1])$ correspond to N-1 samples received prior to sample $X(n)$;

n is the running index of the time coefficients being computed;

N is equal to the number of terms in the equation to the right side of the equal sign;

wherein calculated values of Y define said driving signal.

(27) A method according to claim 26, wherein said step of generating a driving signal further comprises the steps of:

dividing said values of Y by a number previously used to convert initial real number time coefficients to said integer time coefficients; and

discarding any remaining fractional portion of said divided values of Y.

(28) A method according to claim 23, wherein said receiving step comprises the step of reading said series of digital samples from a digital recording medium.

(29) A method according to claim 23, wherein said receiving step comprises the step of reading said series of digital samples from a compressed file.

(30) A method according to claim 23, wherein said receiving step comprises the step of downloading audio sample streams from the Internet.

(31) Apparatus for enhancing a series of digital audio samples comprising:

a device for receiving said series of digital samples,

a digital filter comprising a series of stored time response coefficients, said time response coefficients being mapped to a like number of frequency response coefficients, said frequency response coefficients being spaced at frequency intervals, having phase angles of zero and having amplitudes which are mirrored about a mid frequency, and

a microprocessor for generating a driving signal by convolving said sound samples in real time against said time coefficients.

(32) Apparatus as set forth in claim 31, wherein said receiving device comprises a digital signal reader.

(33) Apparatus as set forth in claim 31, wherein said microprocessor convolves said sound samples in real time against said time response coefficients by repeatedly solving the following equation for Y:

$$Y(n) = A_0X(n) + A_1X(n-1) + \ldots + A_{N-1}X(n-[N-1])$$

where A_0 through A_{N-1} are said stored time coefficients;

$X(n)$ is the most recent sample received;

$X(n-1)$ through $X(n-[N-1])$ correspond to N-1 samples received prior to sample $X(n)$;

n is the running index of the time coefficients being computed;

N is equal to the number of terms in the equation to the right side of the equal sign;

wherein calculated values of Y define said driving signal.

(34) Apparatus as set forth in claim 33, wherein said microprocessor further divides said values of Y by a number previously used to convert real number time coefficients to integer time coefficients, said integer time coefficients defining said stored time coefficients, and said microprocessor further discarding any remaining fractional portion of said divided values of Y.

(35) Apparatus as set forth in claim 31, further comprising a converting device responsive to said driving signal for generating an analog audio signal from said driving signal.

(36) A filter package having two or more parallel digital filters comprising:

a first digital filter comprising a series of digitized first time coefficients stored in a first memory, said time coefficients being mapped to a like number of first frequency coefficients, said first frequency coefficients having either zero phase angles or linearly spaced phase angles and having amplitudes which are mirrored about a mid frequency, and

a second digital filter comprising a series of digitized second time coefficients at least one of which has a value which is different from each of said first time coefficients, said second time coefficients being stored in a second memory and mapped to a like number of second frequency coefficients, said second frequency coefficients having either zero phase angles or linearly spaced phase angles and having amplitudes which are mirrored about a mid frequency.

(37) A filter package as set forth in claim 36, wherein the number of said first time coefficients is equal to or greater than 5 and the number of said second time coefficients is equal to or greater than 5.

(38) A filter package as set forth in claim 36, wherein said first memory and said second memory comprise the same memory component.

(39) Apparatus for enhancing a series of digital audio samples comprising:

a device for receiving said series of digital samples,

a filter package having a first digital filter comprising a series of digitized first time response coefficients stored in a first memory, said time coefficients being mapped to a like number of first frequency coefficients, said first frequency coefficients having either zero phase angles or linearly spaced phase angles and having amplitudes which are mirrored about a mid frequency, and a second digital filter comprising a series of digitized second time response coefficients at least one of which has a value which is different from each of said first time coefficients, said second time coefficients being stored in a second memory and mapped to a like number of second frequency coefficients, said second frequency coefficients having either zero phase angles or linearly spaced phase angles and having amplitudes which are mirrored about a mid frequency, and

a microprocessor for generating a driving signal by convolving said sound samples in real time against either said first time response coefficients or said second time response coefficients.

(40) Apparatus as set forth in claim 39, wherein said first memory and said second memory comprise the same memory component.

(41) Apparatus as set forth in claim 39, further comprising an input device coupled to said microprocessor for selecting one of said first filter and said second filter, said microprocessor generating said driving signal by convolving said sound samples in real time against said first time response coefficients when said first filter is selected and said second time response coefficients when said second filter is selected.

(42) Apparatus as set forth in claim 39, further comprising a converting device responsive to said driving signal for generating an analog audio signal from said driving signal.